

Echo Cancellation Algorithms using Adaptive Filters: A Comparative Study

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Abstract— An adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Adaptive filter finds its essence in applications such as echo cancellation, noise cancellation, system identification and many others. This paper briefly discusses LMS, NLMS and RLS adaptive filter algorithms for echo cancellation. For the analysis, an acoustic echo canceller is built using LMS, NLMS and RLS algorithms and the echo cancelled samples are studied using Spectrogram. The analysis is further extended with its cross-correlation and ERLE (Echo Return Loss Enhancement) results. Finally, this paper concludes with a better adaptive filter algorithm for Echo cancellation. The implementation and analysis is done using MATLAB®, SIMULINK® and SPECTROGRAM V5.0®.

Index Terms— acoustics, echo, echo cancellation, MATLAB®, SIMULINK®, spectrogram, ERLE, cross-correlation

I. INTRODUCTION

FIR and IIR digital filters are designed for applications where the desired filter co-efficients are constants. There are several other digital signal processing applications such as echo cancellation, noise cancellation where the filter co-efficients are variables and cannot be specified a priori. The only way to get the variable filter co-efficients is by an equalizer which has adjustable filter co-efficients, which can be optimized to minimize distortion, on the basis of pre-measured channel characteristics. Such a filter is called an *Adaptive Filter* [1].

An adaptive filter used in acoustic echo cancellation is given in Fig. 1. An acoustic echo is one of the simplest acoustic modelling problems. It happens when the direct signal follows multipath propagation as shown in Fig. 2.

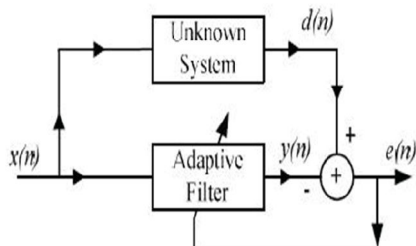


Figure 1. Adaptive filter in Acoustic Echo Cancellation

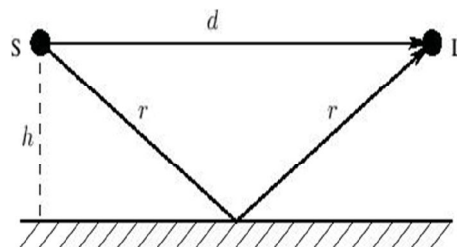


Figure 2. Multipath Propagation

DOI: 01.IJRTET.10.2.14

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As depicted in Fig. 2, direct signal d from the source S at height h reaches the listener L which is followed by the reflected signals r having the magnitude almost same as direct signal. It is referred to as *Echo signal*. It is formed when the direct signal hits the obstacles in the room and gets reflected. Such an echo signal needs to be eliminated or suppressed for better signal perception [2].

For echo cancellation, adaptive filters driven by an error signal are used. Adaptive filters have adjustable filter parameters to minimize the undesired signal by using an adaptive algorithm. There are numerous adaptive algorithms used in an adaptive filter, out of which LMS (Least Mean Square) Algorithm, NLMS (Normalized Least Mean Square) Algorithm, RLS (Recursive Least Square) Algorithm are prominent and widely used.

Spectrogram [3] is the display of the magnitude of the *Short-Time Fourier Transform*. In the spectrogram display, the x-axis represents the time-index and y-axis represents the frequency, whereas the magnitude is represented by the darkness of the plot.

ERLE [4] (Echo Return Loss Enhancement) is defined as the ratio of the power of the desired signal over the power of the residual signal. It is a smoothed measure of the amount (in dB) that the echo has been attenuated. ERLE should stabilize in the interval [-40dB, 30dB] for a good performance. ERLE is used to measure the potential of echo cancellation.

Cross-correlation estimates the similarity between desired signal and echo cancelled signal.

This paper performs acoustical echo cancellation using an adaptive filter driven by LMS, NLMS and RLS algorithm and analyses the echo cancelled/suppressed signals obtained from adaptive filter output using spectrogram, ERLE, cross-correlation and come out with a better algorithm among LMS, NLMS and RLS algorithm for acoustical echo cancellation.

II. LITERATURE REVIEW

This section discusses the literature review of adaptive filters, echo and the process of echo cancellation.

A. Adaptive Digital Filters

Adaptive filters are self-learning filters, whereby an FIR or IIR filter is designed based on the characteristics of input signals to adapt its environment. The environment will be defined by the input signal $x(n)$ and desired signal $d(n)$. Adaptive filters have self-regulation and tracking capabilities. An adaptive filter finds its essence in applications such as Echo Cancellation, Noise Cancellation, System Identification and many others. A basic adaptive filter was first invented at AT&T Bell Labs. Since its inception, several adaptive filter algorithms were designed and honed. A few algorithms include LMS, NLMS and RLS algorithms. These algorithms were designed to anticipate the signal which would inevitably re-enter the transmission path and cancel it out.

A few adaptive filter algorithms are discussed below:

1. *LMS (Least Mean Square) Algorithm* [5] - It is a stochastic gradient-based algorithm introduced by Bernard Widrow and Ted Hoff which uses gradient vector of the filter tap weights in order to converge on the optimal Wiener solution. In each iteration of the algorithm, the filter taps weights are updated as per Equation (3) where $w(n)$ represents the adaptive filter weight vector at time n , $x(n)$ represents time-delayed input signal samples, $e(n)$ represents error signal to be minimized and μ represents step size or convergence factor.

$$\text{Output, } y(n) = w^h x(n) \quad (1)$$

$$\text{Error, } e(n) = d(n) - y(n) \quad (2)$$

$$\text{Weight, } w(n+1) = w(n) + \mu x(n)e(n) \quad (3)$$

If μ is chosen to be very small then the algorithm converges very slowly. A large value of μ may lead to a faster convergence but the adaptive filter becomes less stable around the minimum value and its output diverges.

2. *NLMS (Normalized LMS) Algorithm* [5] - In LMS algorithm, when the values of μ is large, the algorithm experiences a gradient noise amplification problem. This problem is solved by NLMS algorithm. The

correction applied to weight vector $w(n)$ at iteration $n+1$ is normalized with respect to the squared Euclidian norm of the input vector $x(n)$ at iteration n . The NLMS algorithm can be viewed as a time-varying step-size algorithm, calculating the convergence factor μ as in Equation (4).

$$\mu(n) = \frac{\alpha}{c + \|x(n)\|^2} \quad (4)$$

In Equation (4), α is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition $0 < \alpha < 2$, and c is the constant term for normalization, which is always less than 1.

The filter weights using NLMS algorithm are updated as given in Equation (5).

$$w(n+1) = w(n) + \frac{\alpha}{c + \|x(n)\|^2} e(n)x(n) \quad (5)$$

3. *RLS (Recursive Least Square) Algorithm* [6] – This algorithm attempts to minimize the cost function in Equation (6). In Equation (6), $k=1$ is the time at which the RLS algorithm commences and λ is a small positive constant very close to, but smaller than 1. With values of $\lambda < 1$ more recent input samples, this results in a scheme that places more emphasis on recent samples of observed data and tends to forget the past.

$$\zeta(n) = \sum_{k=1}^n \lambda^{n-k} e_n^k(k) \quad (6)$$

When compared to LMS algorithm, RLS algorithm offers a faster convergence and lower error at steady state. But, this RLS algorithm is much more computationally complex and if proper design procedures are not followed, RLS algorithm may diverge away resulting in instability.

B. Echo [7]

Echoes are simply generated by delay units. The direct sound and a single echo appearing after R sampling periods later can be generated by the FIR filter as shown in Fig. 3.

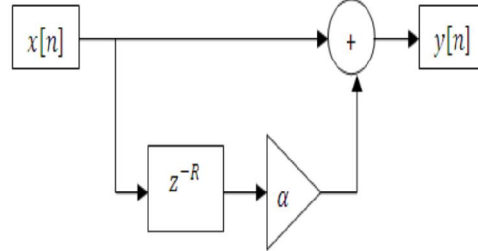


Figure 3. Echo filter

The transfer function of the echo filter is given by Equation (7).

$$H(Z) = 1 + \alpha Z^{-R}, |\alpha| < 1 \quad (7)$$

In the above transfer function, the delay parameter R denotes the time the sound wave takes to travel from the sound source to the listener after bouncing back from the reflecting wall, whereas the parameter, α with $|\alpha| < 1$, represents the signal loss caused by propagation and reflection.

There are two types of Echo – *Acoustic Echo* and *Hybrid Echo*. Hybrid Echo is generated in PSTN Network.

C. Echo Cancellation

Echo cancellation is the process of removing echo signals from a voice communication system in order to achieve quality audio perception. The development of echo reduction began in the late 1950s, and continues today as new integrated landline and wireless cellular networks put additional requirement on the

performance of echo cancellers. Echo cancellation involves in first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it is removed by 'subtracting' it from the transmitted or received signal. This technique is usually implemented on DSP's using adaptive filters.

III. DIGITAL IMPLEMENTATION USING MATLAB-SIMULINK®

This section describes the digital implementation of various echo cancellation algorithms using MATLAB-SIMULINK® V7.5. We employ SIMULINK® *Signal Processing Toolbox* and the common blocks used were *From Multimedia File*, *Delay*, *Signal to Workspace*, *Gain*, *Sum*, *To Audio Device*, *LMS Filter*, *NLMS Filter* and *RLS Filter*.

A. Echo Model [7]

This model generates Echo signal for as given input signal. Here we use PCM 16 bit signed, 352 kbps, 22050Hz speech signal as input. The Echo – Simulink model is represented in Fig. 4.

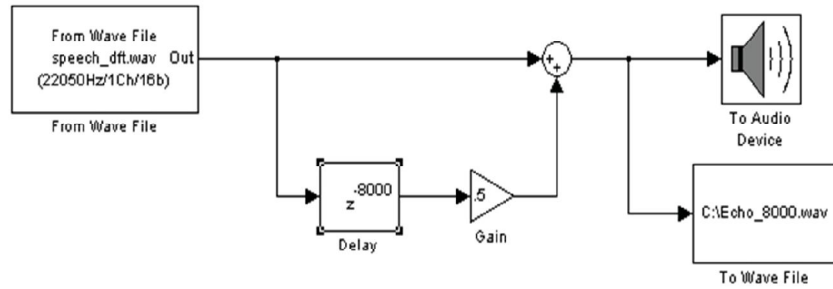


Figure 4. Echo – Simulink model

B. Echo Cancellation Model

We implement Echo Cancellation model using LMS, NLMS and RLS algorithms which are shown in Fig. 6, Fig. 7, and Fig. 8 respectively. Fig. 5 represents LMS filter Simulink block.

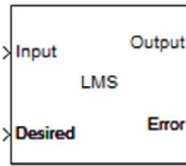


Figure 5. LMS Filter – Simulink Block model

The above block has the below input and output ports.

Input Port	:	Signal + its Echo
Desired Port	:	Desired Audio signal
Output Port	:	Echo cancelled/suppressed signal
Error Port	:	Difference between desired signal and adaptive filter output

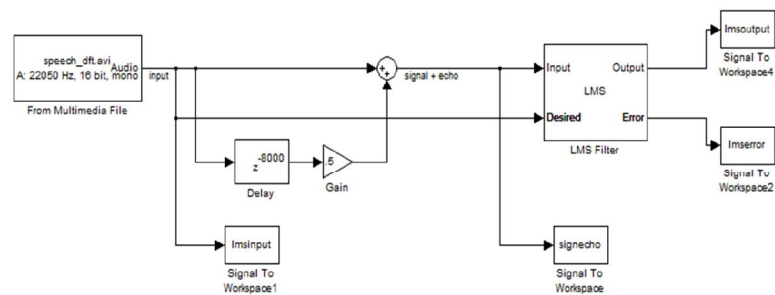


Figure 6. LMS Echo Cancellation – Simulink model

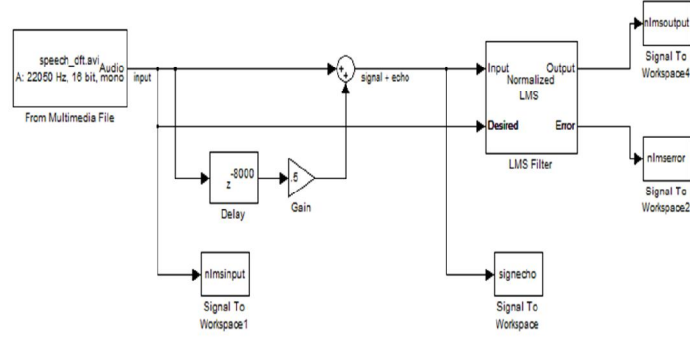


Figure 7. NLMS Echo Cancellation – Simulink model

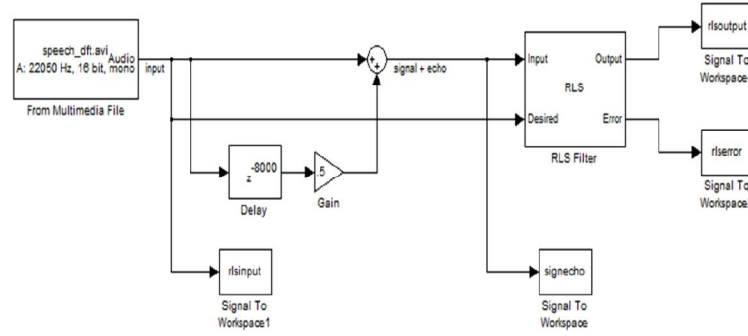


Figure 8. RLS Echo Cancellation – Simulink model

C. Experimental/Implementation Procedure:

STAGE 1:

- 1) Initially, a speech signal without any echo components is used as desired/ideal signal. It is a PCM (Pulse Code Modulation) signed 16-bit, 352kbps, 22050Hz, 1 channel speech signal.
- 2) Echo Model is implemented using Simulink and the value of α is kept constant at 0.5 and the value of R (delay) is kept constant at 8000 samples throughout the experiment. Generated signal is an echo signal.
- 3) The desired signal and signal+Echo will be used as inputs to LMS Echo Canceller.
- 4) Simulation is run for 8 seconds and the Output port of LMS block gives the echo cancelled signal. Error Port of LMS block gives the difference between desired and LMS output.
- 5) Desired signal, Signal+Echo, Output signal and Error signal are saved in the workspace for further analysis.
- 6) Spectrogram Plots are obtained for desired and output signals using *SPECTROGRAM V.5.0* Tool [8]
- 7) The same procedure is repeated for NLMS and RLS Echo Canceller Algorithms

STAGE 2:

- 1) We use the concept of ERLE [4] (Echo Return Loss Estimation) to measure the potential of Echo cancellation. It is defined as the ratio of the power of the desired signal over the power of the residual signal. The expression to determine ERLE is given in Equation (8).

$$ERLE = -10 \log_{10} \frac{E(d^2(n))}{E(e^2(n))} \text{ dB} \quad (8)$$

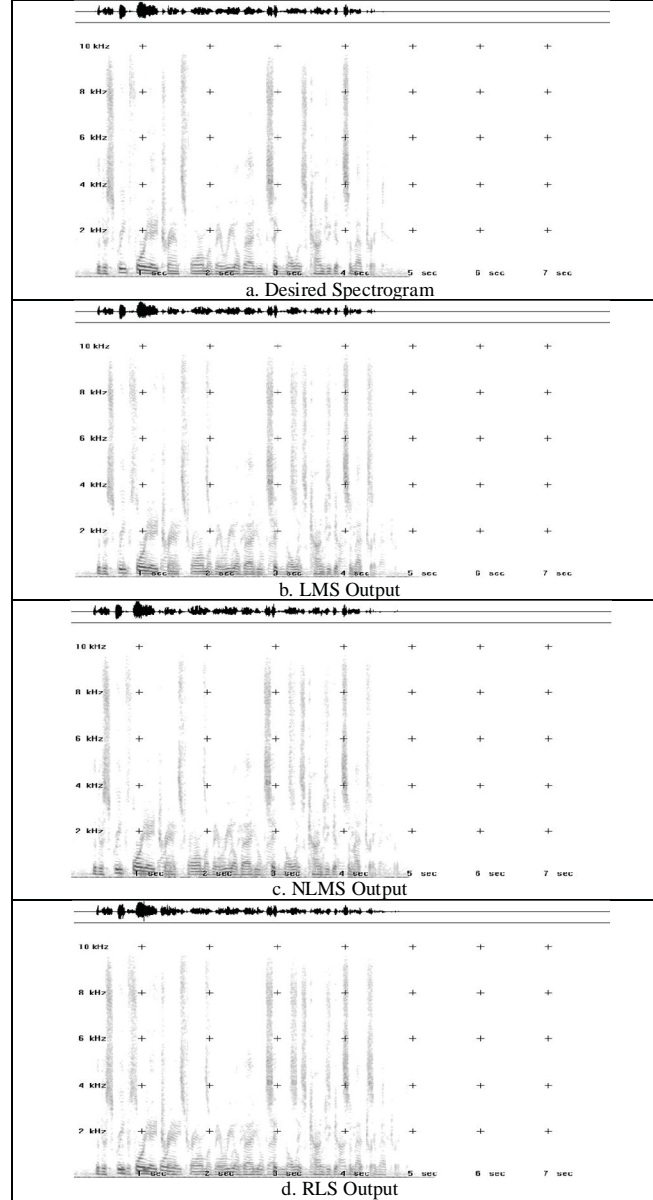
- 2) It is a smoothed measure of the amount (in dB) that the echo has been attenuated. ERLE should stabilize in the interval [-40dB, 30dB] for a good performance

STAGE 3:

- 1) To determine the amount of time shift between desired and echo cancelled signal, we use the concept of cross-correlation

- 2) Cross correlation is calculated between desired signal and echo cancelled output obtained from LMS, NLMS and RLS adaptive filter algorithms.
- 3) Ideally, the time shift between desired and echo cancelled signal should be minimal; and the amplitude of time shifted signal should be very less for better audio perception

TABLE I SPECTROGRAM PLOT



IV. IMPLEMENTATION ANALYSIS

This section discusses the analysis of Echo cancellation algorithm implemented using Spectrogram, ERLE and Cross-Correlation.

TABLE II
ERLE PLOT

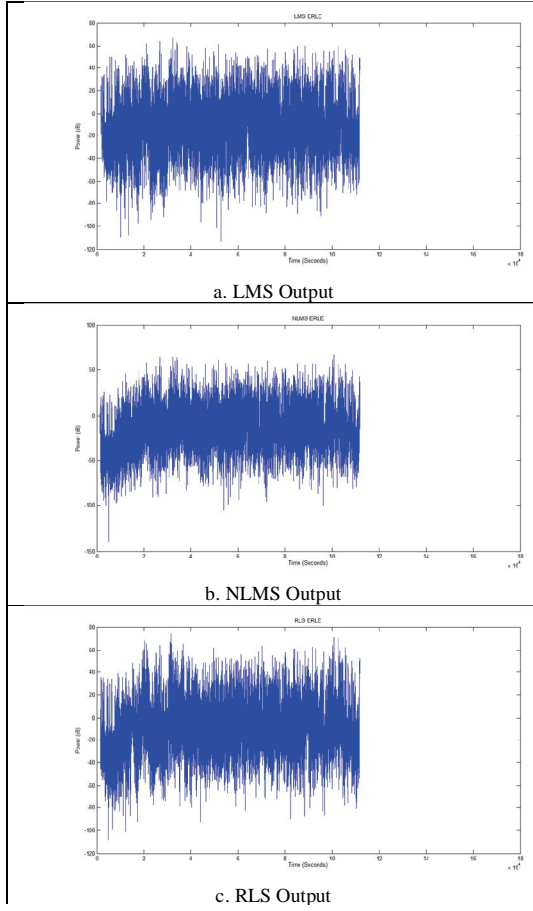
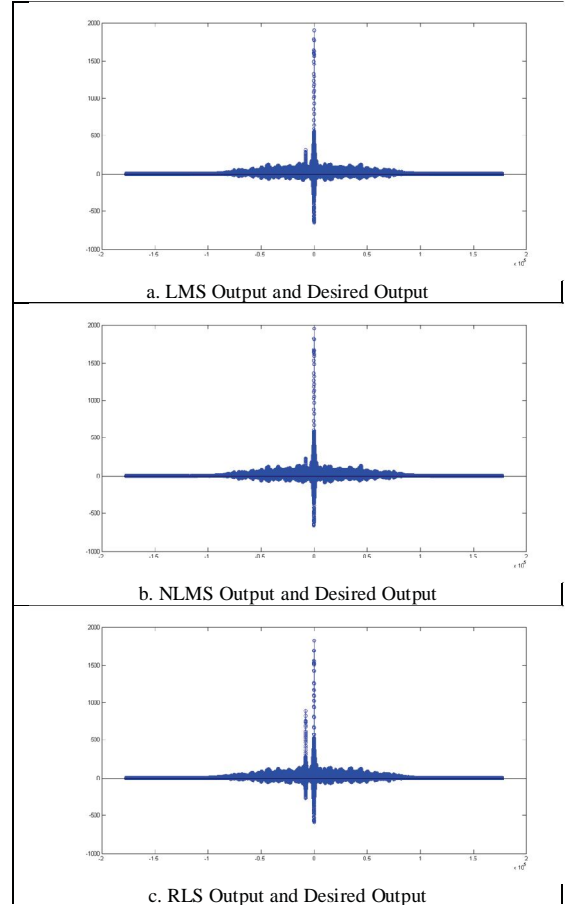


TABLE III
CROSS-CORRELATION PLOT



Spectrogram Analysis – For LMS, NLMS and RLS echo canceller output samples, spectrograms were determined and the plots are represented in Table I. From the spectrogram plots we observe that the NLMS cancels the echo signals to a maximum extent and RLS cancels the echo signals to a minimum extent whereas in LMS algorithm, echo signals are cancelled out moderately.

ERLE Analysis – As discussed earlier, ERLE measures the potential of echo cancellation. It is calculated as per the Equation (8). The ERLE plots for LMS, NLMS, and RLS algorithms are represented in Table II. For LMS algorithm, ERLE value lies in the range [-65dB, 50dB]. For NLMS algorithm, ERLE value lies in the range [-60dB, 40dB] and for RLS algorithm, ERLE value lies in the range [-80dB, 60dB]. But, ERLE value has to stabilize in the range [-40dB, 30dB] for better performance. Hence, NLMS algorithm offers better performance when compared to LMS and RLS.

Cross-Correlation Analysis – It is used to determine the time-shift between two signals. The cross-correlation plots of LMS, NLMS and RLS algorithms are represented in Table III.

From Table 3 we observe that the amplitude of time shifted signal (Echo signal) is minimum for NLMS and maximum for RLS, whereas the amplitude of LMS between the two. Hence, NLMS algorithm offers better echo cancellation.

V. CONCLUSION

Considering the Spectrogram analysis, cross correlation and ERLE results of three adaptive filter algorithms, this paper concludes that the NLMS algorithm is best suited for echo cancellation. NLMS algorithm provides

better ERLE stability in the range [-40dB, 30dB]. Also the amplitude of time shifted in Cross Correlation plots is minimum for NLMS algorithm.

Listening tests indicate that the perceived temporal quality or texture is better for NLMS, followed by LMS and RLS. This is also evident from Spectrogram, ERLE and Cross-Correlation plots.

This paper also discusses the implementation of LMS, NLMS and RLS adaptive filter algorithms for echo cancellation in a concert hall and it brings out the difference between LMS, NLMS and RLS algorithms. Finally, it performs a better analysis of echo cancellation algorithms considering Spectrogram, ERLE and cross-correlation.

This paper concludes with the NLMS adaptive filter algorithm to be a better algorithm amongst LMS, NLMS and RLS for echo cancellation.

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